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Study and Analysis of FIR Filters with Various Filtering Techniques

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Abstract— In digital control system, interference is mixed in the input signal which has a great influence on the performance of the system. Therefore, processing of input signal has to be done to get useful signal. Finite impulse response (FIR) filter plays an important role in the processing of digital signal. Designing the FIR filter by MATLAB can simplify the complicated computation in simulation and improve the performance. By using the methods of window function, frequency sampling and convex optimization techniques, the design of FIR filter has been processed by MATLAB. In the view of the designed program of MATLAB and we can get the amplitude-frequency characterization. FIR digital filters have been designed to process the input signal based on the MATLAB function, the filtering effect of different digital filters is analyzed by comparing the signal's amplitude-frequency diagrams which have been generated. The experimental results show that the FIR filters designed are effective.

Keywords— FIR filter, MATLAB, window function, frequency sampling, optimization, amplitude-frequency characterization

I. INTRODUCTION

The digital filter is a discrete system, and it can do a series of mathematic processing to the input signal, and therefore obtain the desired information from the input signal. The transfer function for a linear, time-invariant, digital filter is usually expressed as

$$H(z) = \frac{\sum_{j=0}^{M} b_j z^{-j}}{1+\sum_{i=1}^{N} a_i z^{-i}}$$

Where a_i and b_j are coefficients of the filter in Z-transform.

There are many kinds of digital filters, and also many different ways to classify them. According their function, the FIR filters can be classified into four categories, which are low pass filter, high pass filter, band pass filter, and band stop filter. According to the impulse response, there are usually two types of digital filters, which are finite impulse response (FIR) filters and infinite impulse response (IIR) filters. According to the formula above, if a_i is always zero, then it is a FIR filter, otherwise, if there is at least one none-zero a_i, then it is an IIR filter. The three basic arithmetic units needed to design a digital filter are: the adder, the delay, and the multiplier.

The following are the steps for designing a digital filter:

1. Make sure of the property of a digital filter according to the given requirements.
2. Use a discrete linear time-invariant system function to approach to the properties.
3. Make use of algorithms to design the system function.
4. Use a computer simulation or hardware to achieve it.

II. FIR FILTER

The finite impulse response (FIR) filter is one of the most basic elements in a digital signal processing system. It can guarantee a strict linear phase frequency characteristic and amplitude frequency characteristic. Besides, the unit impulse response is finite, therefore FIR filters are stable system. The FIR filter has abroad application in many fields, such as telecommunication, image processing, and so on.

The system function of FIR filter is:

$$H(z) = \sum_{n=0}^{L-1} h[n]\, z^{-n},$$

Where L is the length of the filter, and h[z] is the impulse response.

III. IIR FILTER

The infinite impulse response (IIR) filter is recursive structure, and it has a feedback loop. The precision of amplitude frequency characteristic is very high, and IIR filters are not linear phase.The input and the output signals are related by

$$y(n) = \sum_{k=0}^{\infty} h(k) x(n-k)$$

where h (k) is the impulse response sequence where k = 0,1,……..,

from this equations we see that, for IIR filters, the impulse response is of infinite duration.
The IIR filtering equation is expressed in a recursive form

$$y(n) = \sum_{k=0}^{\infty} h(k)x(n-k) = \sum_{k=0}^{M} b_k x(n-k) - \sum_{k=1}^{N} a_k y(n-k)$$

where the a_k and b_k are the coefficients of the filter. From equation we note that, the current output sample, y (n) is a function of past values of output and the present and past input samples, that is the feedback system. The equation reduces to the FIR equation when the b_k are set to zero and we note that in the FIR filter current output sample, y(n) is a function only of past and present values of input sample.

IV. COMPARISON OF FIR AND IIR FILTERS

Table 1 Comparison of FIR and IIR Filters

S. No.	FIR	IIR
1.	FIR filters are stable	IIR filters are unstable
2.	Linear phase Response	Difficult to control phase
3.	Can be used for higher order	Only for lower order
4.	Large computations	Lesser Computations
5.	They are Non Recursive Filters	They are Recursive Filters
6.	Storage units are more	Storage units are lesser
7.	FFT technique can be used	FFT cannot be used

V. SIMULATION OF ELECTRONIC COMMUNICATION SYSTEM

The Concept of Telecommunications and Electronic systems simulation

System simulation technology refers to computer simulation technology, which developed since 1970 combining modern computers and simulation software. Computer simulation has high precision, versatility, good repeatability, rapid modelling and low cost advantages. Especially in recent years, MATLAB, a scientific computing and system simulation language, is developed rapidly [16]. It's much more convenient to use and operate other than traditional C or C++ language. So-called electronic communication system simulation is the use of the computer on the physical model or mathematical model of actual electronic communication systems testing, analysis and study of this model test on an actual system performance and operating state. When a pilot study in actual electronic communication system is more difficult or impossible to achieve, the simulation technology has become an inevitable choice.

Steps of computer simulation

(1) Analyze the simulation system
(2) Build a system mathematical model
(3) Collect data
(4) Establish simulation model on computer
(5) Verify the simulation model
(6) Confirm the simulation model
(7) Design simulation experiment
(8) Run the computer simulation model
(9) Analyse results of the simulation

VI. DESIGN OF FIR FILTER

It is necessary to specify pass band, stop band, and transition band when designing a frequency-selective filter. In pass band, frequencies are needed to be passed un - attenuated. In stop band, frequencies need to be passed attenuated. Transition band contain frequencies which are lying between the pass band and stop band. Therefore, the entire frequency range is split into one or even more pass bands, stop bands, and transition bands. In practical, the magnitude is not necessary to be constant in the pass band of a filter. A small amount of ripple is usually allowed in the pass band. Similarly, the filter response does not to be zero in the stop band. A small, nonzero value is also tolerable in the stop band.

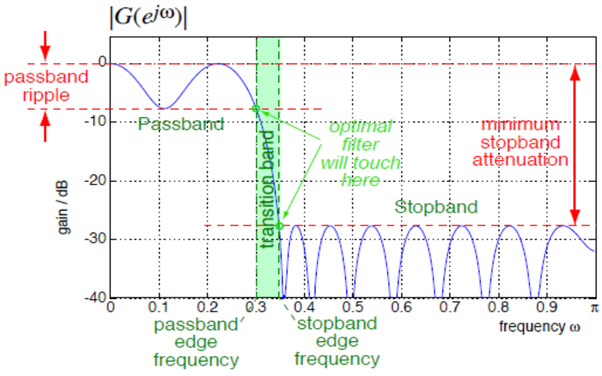

Fig.1 Amplitude-Frequency characteristic of low pass filter

The transition band of the filter is between the pass band and the stop band. The frequency ω_p denotes the edge of the pass band, and the band-edge frequency ω_s defines the edge of the stop band. So, the difference of ω_s and ω_p is the width of the transition band, i.e. $\omega_t= \omega_s- \omega_p$ The ripple in the pass band of the filter is denoted as δ_p, and the magnitude of the filter varies from $1-\delta_p$ to $1+ \delta_p$. δ_s is the ripple in the stop band. Usually we use a logarithmic scale to show the frequency response, hence, the ripple in the pass band is $20\log10\delta_p$ dB, and the ripple in the stop band is $20\log10\delta_s$dB. The characteristics of digital filters are often specified in the frequency domain. The amplitude – frequency response of FIR and IIR filter is often specified in the form of a tolerance scheme as given in the figure 1. Referring to the figure above, the following parameters are of interest:

1. Peak pass band deviation (or ripples)
2. Stop band deviation
3. Pass band edge frequency
4. Stop band edge frequency

VII. VARIOUS TECHNIQUES OF FILTERS

The various filter techniques used to remove unwanted signal i.e. interference to get a meaningful signal are:

1. Window Method
2. Frequency Sampling Method
3. Optimum Equiripple Method

WINDOW METHOD

In this method, a truncated ideal low pass filter with a certain bandwidth is generated, and then a window is chosen to get certain stop band attenuation. The length of filter ‘L’ can be adjusted to meet a specified roll-off rate in the transition band. The windowed, truncated low pass filters is considered and then other kind of filters, like high pass, band pass, and band stop filters can also be achieved by several techniques [17]. Any finite-length of the ideal low pass impulse response may be considered as the product of the infinite-length low pass impulse response may be considered as the product of the infinite-length low pass impulse response and a window function *W*, which has a finite number of contiguous nonzero-valued amples.

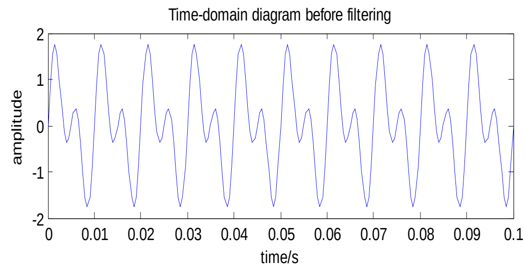

Fig.2 Time-domain before filtering

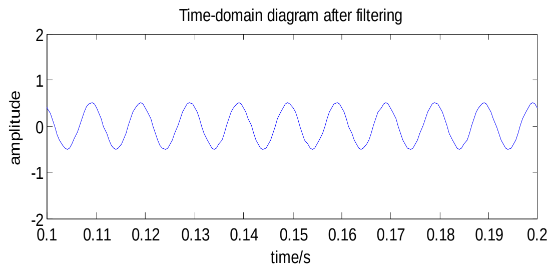

Fig.3 Time –domain after filtering

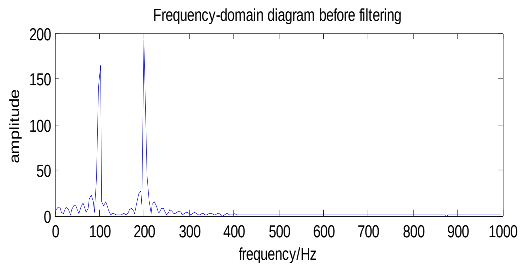

Fig.4 Frequency-domain before filtering

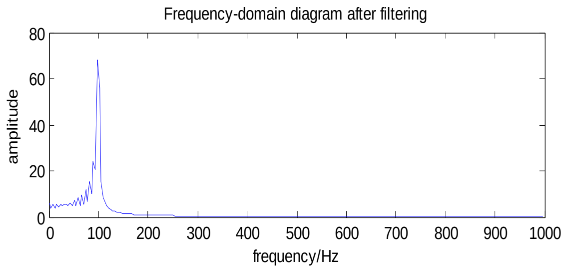

Fig.5 Frequency –domain after filtering

Comparing Figures 2,3,4,5 it is easy to see that the input signal is made up of two superposition signals with different frequencies. The pass band is 0-100 Hz, and stop band starts from 170Hz, and there are two frequencies of 100Hz and 200Hz needing to be filtered. The signal with frequency of 100Hz which is in the range of pass band is kept, while the signal with frequency of 200Hz which is in the range of stop band is filtered. The minimum stop band attenuation values for Hamming window is 53dB, which is greater than 50dB, and this result exactly meets the requirements.

Frequency Sampling

The frequency sampling method will work in following way, we start in the frequency domain, and sample the desired frequency response H(ejΩ) with N evenly-spaced samples instead of a continuous frequency, and get $H_d(k)= H_d(ej\Omega) \mid \Omega=2\pi k/N$, (k=0,1,…, N-1). Then, let $H(k)= H_d(k)= H_d(ej\Omega)|\Omega=2\pi k/N$, we get the unit impulse response, h(n)=IDFT[H(k)], where IDFT is Inverse Discrete Fourier Transform. The inverse DFT then yields an impulse response which will lead to a filter whose frequency response the same as that of the specification exactly at the location of the frequency samples. The advantage of this method is that we can design filters directly in the frequency domain, but the disadvantage is that the sampling frequency can only be integer times of 2π/N, and we cannot ensure a random cutoff frequency.

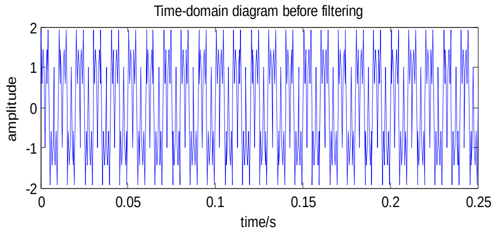

Fig.6 Time-domain before filtering

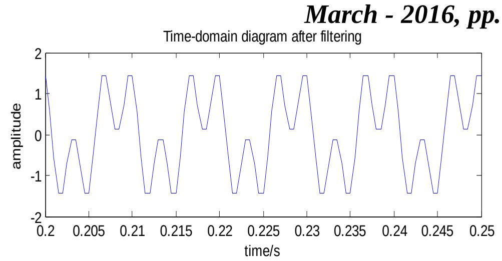

Fig.7 Time –domain after filtering

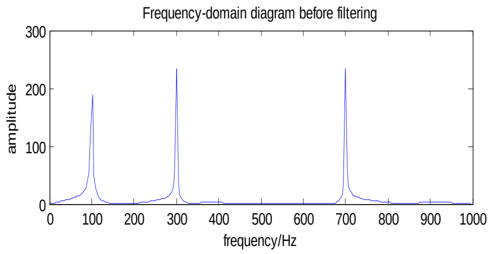

Fig.8 Frequency-domain before filtering

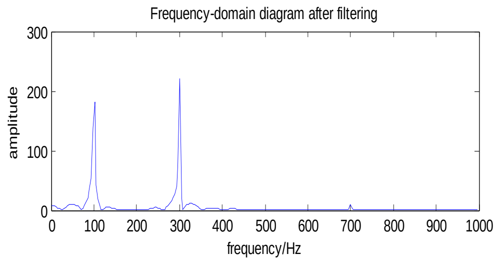

Fig.9 Frequency –domain after filtering

From the figures 6,7,8,9 above, we can get the information that the input signal is made up of three superposition signals with distinct frequencies. The sampling frequency is 2000Hz, and the pass band is from 0 to 500Hz, so signals with the frequencies of 100Hz and 300Hz are kept, while that of 700Hz is filtered.

Optimum Equiripple Method

Historically, the window function method was the first method for designing linear-phase FIR filters. The frequency sampling method and optimized equiripple method were developed in the 1970s and have become very popular since then.Lacking precise control of the specified frequencies, like ω_p and ω_s, is the most serious disadvantage of the window function method in the design of a low pass FIR filter. The frequency sampling method is better than the window method in the aspect that the real-valued frequency response characteristics $H_r(\omega)$ is specified, which can be either zero or unity at all frequencies, except the transition band. The Chebyshev approximation method offers completely control of the filter requirements. As a result, this method is more preferable than the other two. It is based on the Remez exchange algorithm, which minimizes the error with respect to the max-norm.

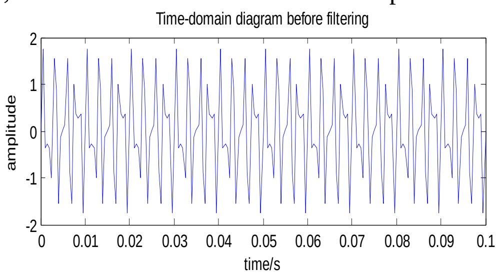

Fig.10 Time-domain before filtering

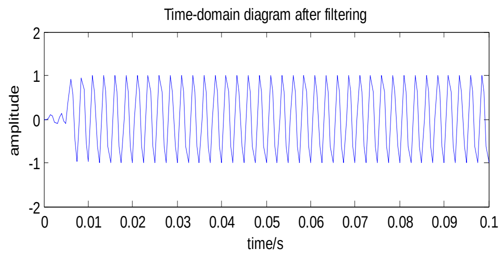

Fig.11 Time –domain after filtering

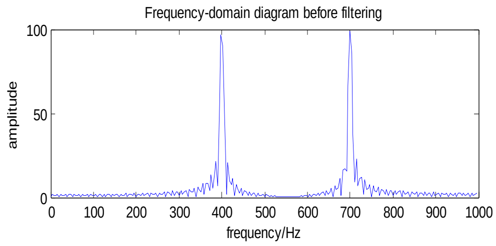

Fig.12 Frequency-domain before filtering

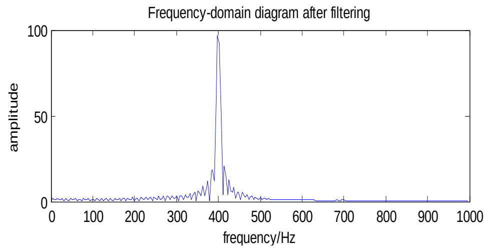

Fig.13 Frequency –domain after filtering

From the figures above i.e. 10,11,12,13, it is easy to see that the input signal is made up of two superposition signals with distinct frequencies. The pass band is from 0 to 500Hz, and the stop band is from 500 to 1000Hz. After filtering, the signal with frequency of 400Hz is kept, while the signal with frequency of 700Hz which is in the stop band is filtered.

VIII. CONCLUSION

Out of all FIR Filtering techniques, OPTIMUM EQUIRIPPLE METHOD is the best method for filtering, because of:

1. Minimizes maximum ripple
2. Precise control of the critical filter frequencies
3. Can be easily designed on computer that's why called computer aided design

4. Ideal linear phase design
5. Approximate error between desired frequency response & actual frequency response is spread evenly across pass band and stop band
6. Sharpest FIR filter

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